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EXAMINER
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ART UNIT	PAPER NUMBER
2644	13

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Please find below and/or attached an Office communication concerning this application or proceeding.

	Application No.	Applicant(s)
Office Action Summary	09/060,825	BRENNAN, ROBERT
	Examiner	Art Unit
	Con P. Tran	2644

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE <u>3</u> MONTH(S) FROM
THE MAILING DATE OF THIS COMMUNICATION.
- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed
  after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133).
- Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any
  earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on <u>22 November 2002</u> .

2a)☐ This action is **FINAL**.   2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is
closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1-30* is/are pending in the application.

    4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-30* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

    Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

11)☐ The proposed drawing correction filed on _____ is: a)☐ approved b)☐ disapproved by the Examiner.

    If approved, corrected drawings are required in reply to this Office action.

12)☐ The oath or declaration is objected to by the Examiner.

Priority under 35 U.S.C. §§ 119 and 120

13)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

    a)☐ All b)☐ Some * c)☐ None of:

      1.☐ Certified copies of the priority documents have been received.

      2.☐ Certified copies of the priority documents have been received in Application No. _____ .

      3.☐ Copies of the certified copies of the priority documents have been received in this National Stage
        application from the International Bureau (PCT Rule 17.2(a)).

    * See the attached detailed Office action for a list of the certified copies not received.

14)☐ Acknowledgment is made of a claim for domestic priority under 35 U.S.C. § 119(e) (to a provisional application).

    a) ☐ The translation of the foreign language provisional application has been received.

15)☐ Acknowledgment is made of a claim for domestic priority under 35 U.S.C. §§ 120 and/or 121.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)		4) ☐ Interview Summary (PTO-413) Paper No(s). _____ .
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)		5) ☐ Notice of Informal Patent Application (PTO-152)
3) ☐ Information Disclosure Statement(s) (PTO-1449) Paper No(s) _____ .		6) ☐ Other: .

DETAILED ACTION

Claim Rejections - 35 USC § 103

1.   The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all

obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

2.   **Claims 1, 25, 6, 12-14** are rejected under 35 U.S.C. 103(a) as being

unpatentable over Sheikhzadeh, Sameti, Deng and Brennan ("Comparative

Performance of Spectral Subtraction and HMM-Based Speech Enhancement Strategies

with Application to hearing Aid Design", Acoustics, Speech, and Signal Processing,

1994. ICASSP-94. , 1994 IEEE Int. Conf.) in view of Händel (Händel; PCT WO

96/24128), and further in view of Eatwell U.S. Patent 5,742,694.

Regarding **claim 1**, Sheikhzadeh, Sameti, Deng and Brennan teaches a method

of reducing noise in an input, the input signal containing speech, and having a signal to

noise ratio, the method comprising the steps: detect the absence of speech; in the

absence of speech determining a noise magnitude spectral estimate ($| V_t^\wedge(w) |$), (page I-

13, left-hand column, paragraph 2 - right-hand column, paragraph 2).

However, Sheikhzadeh, Sameti, Deng and Brennan does not explicitly show

(1) detecting the presence of speech;

(3) in the presence of speech comparing the magnitude of the audio signal

| X(f)|, to the noise magnitude spectral estimate (| N^(f) |);

(4) calculating an attenuation function H(f) (16) from the magnitude

spectrum of the audio signal | X(f)| (130) and the noise magnitude spectral estimate

|N^(f) | the attenuation function (H(f) being dependent on the signal to noise ratio; and

(5) modifying (see page 17, lines 24-30) the input signal by the attenuation

function H(f) (16) to generate an noise reduced signal (see page 18, lines 1-2).

Thus one of ordinary skill would have been motivated to seek a method of

reducing noise in an input in order to provide a method of an actual working

arrangement taught by Sheikhzadeh, Sameti, Deng and Brennan. Such method would

have been any known method for use in noise reduction such as one of Händel in the

same field of endeavor.

Händel teaches a method of reducing noise (see page 2, lines 10-12) in an input,

the input signal containing speech (see page 2, lines 3-6), and having a signal to noise

ratio (SNR; see page 16, lines 5-6), the method comprising the steps (see Fig. 1 and

Fig. 7):

(1) detecting the presence and absence of speech (see page 4, lines 1-5; and

page 5, lines 22-28);

(2) in the absence of speech (see page 5, lines 13-15), determining a noise

magnitude spectral estimate (| N^(f) |), (140, see Fig. 7);

(3) in the presence of speech comparing the magnitude of the audio signal | X(f)|, (130) to the noise magnitude spectral estimate (| N^(f) |), (see page 5, lines 15-21);

(4) calculating (see page 16 lines 1-page 17, lines 24) an attenuation function H(f) (16) from the magnitude spectrum of the audio signal | X(f)| (130) and the noise magnitude spectral estimate | N^(f) (140) the attenuation function (H(f)) being dependent on the signal to noise ratio (see page 16, lines 1-10); and

(5) modifying (see page 17, lines 24-30) the input signal by the attenuation function H(f) (16) to generate an noise reduced signal (see page 18, lines 1-2);

in order to provide a better noise reduction without sacrificing audible quality (page 2, lines 12-13).

Therefore, it would have been obvious to one of ordinary skill in the art, at the time the invention was made to include within the Sheikhzadeh, Sameti, Deng and Brennan a method of noise reduction system as taught by Händel in order to provide a better noise reduction without sacrificing audible quality as suggested by Händel in page 2, lines 12-13.

However, Händel does not explicitly show in modifying a input signal, wherein there is no substantial modification to the input signal for very low and for very high signal to noise ratios.

Thus one of ordinary skill would have been motivated to seek a method for modifying a input signal, wherein there is no substantial modification to the input signal for very low and for very high signal to noise ratios in order to provide a method of an actual working arrangement taught by Händel. Such method would have been any

known method for use in noise reduction such as one of Eatwell in the same field of

endeavor.

Eatwell teaches a noise reduction filter in which the gains are adjusted according

to estimates of the signal and noise contents of the signal components. When a signal

component has a high signal-to-noise ratio, the gain should be set close to unity to

allow the signal component to pass. When a signal component has a low signal-to-

noise ratio, it is desirable to reduce the level of the component so as to reduce the

noise in the output signal. Similar gain elements are used in the spectral subtraction

method (col. 8, line 63 – col. 9, line 3) in order to provide a noise reduction filter with

low computation and memory requirements (col. 3, lines 26-27).

Therefore, it would have been obvious to one of ordinary skill in the art, at the

time the invention was made to include within the Händel a noise reduction filter to

generate a noise reduced signal using signal having only a high signal-to-noise ratio,

and signal having only a high signal-to-noise ratio; there is no substantial modification

to the input signal for very low and for very high signal to noise ratios; for the purpose

of reducing the noise in the output signal as taught by Eatwell in col. 8, line 63 – col. 9,

line 3 in order to provide a noise reduction filter with low computation and memory

requirements as suggested by Eatwell in column 3, lines 26-27.

Regarding **claim 25,** Händel teaches a method as claimed in claim 1, wherein

the square of the speech magnitude spectral estimate | S(f)|, (Φ_s (w)) is determined by

subtracting the square of the of the noise magnitude spectral estimate N^(f) (i.e. ,(Φ_v

(w))) from the square of the magnitude spectrum of the input signal $| X(f) |$ ((Φ_x(w); see

page 3, lines12-14).

Regarding **claim 6**, Händel further teaches a method of reducing noise (see

page 2, lines 10-12) in an input, audio signal containing speech (see page 2, lines 3-6),

wherein (see page 7, Table 2, line 8) the attenuation function (i.e. $\hat{H}_{\delta PS}$(w) is calculated

in accordance with the following equation:

$$H(f) = \{ (| X(f) |^2 - \beta | N^{\cdot}(f) |^2) / | X(f) |^2) \}^\alpha$$

Where: $H(f) = \hat{H}_{\delta PS}$(w) , attenuation function (see page 7, line 2-3)

 $X(f) = \Phi^{\wedge}_x$(w) , magnitude spectrum of the input audio signal

 (see page 3, line 26-27)

 $N^{\cdot}(f) = \Phi^{\wedge}_v$(w) , noise magnitude spectral estimate

 (see page 4, line 3-4)

 β $= \delta$, oversubtraction factor (see page 26, lines 6-7)

 α $= \frac{1}{2}$, an attenuation rule

Regarding **claim 12**, Sheikhzadeh, Sameti, Deng, Brennan in view of Händel,

and further in view of Eatwell discloses a method of noise reduction as claimed in claim

1. However, the reference does not explicitly include remotely turning noise suppression

on and off.

Nevertheless, as would have been well known in the art at the time the invention

was made, such remote control is conventional for turning an electronics device on and

off. Accordingly, it would have been obvious to one of ordinary skill in the art, at the time

the invention was made to include remotely turning noise suppression on and off

because such method is conventional.

Regarding **claim 13**, the Händel reference discloses a method of noise reduction.

However, the reference does not explicitly include automatically disabling noise reduction

in the presence of very light noise or extremely adverse environments.

Nevertheless, as would have been well known in the art at the time the invention

was made, such specifications are required in order to preserve battery power and to

protect user from extremely loud environments. Accordingly, it would have been obvious

to one of ordinary skill in the art, at the time the invention was made to include

automatically disabling noise reduction in the presence of very light noise or extremely

adverse environments because such specifications would preserve battery power and

to protect user from extremely loud environments.

Regarding **claim 14**, Händel teaches a method of reducing noise (see page 2,

lines 10-12) in an input, audio signal containing speech. The method includes detecting

speech with a modified auto-correlation (page11, lines 14-15).

3.   **Claims 2-5, 21-24, 26-30** are rejected under 35 U.S.C. 103(a) as being

unpatentable over Sheikhzadeh, Sameti, Deng and Brennan ("Comparative

Performance of Spectral Subtraction and HMM-Based Speech Enhancement Strategies

with Application to hearing Aid Design", Acoustics, Speech, and Signal Processing,

1994. ICASSP-94. , 1994 IEEE Int. Conf.) in view of Händel (PCT WO 96/24128) in

view of Eatwell U.S. Patent 5,742,694, further in view of Lindemann et al. (5,479,522).

Regarding **claim 2**, "Sheikhzadeh, Sameti, Deng, Brennan" in view of Händel, and

further in view of Eatwell teaches a method as claimed as claim 1. However

"Sheikhzadeh, Sameti, Deng, Brennan" in view of Händel, and further in view of Eatwell

does not explicitly show the method comprising the steps of:

(6) supplying the input signal to an amplification unit;

(7) providing the noise reduced signal to a compression circuit, which generates

a control signal for the amplification unit;

(8) controlling the amplification unit with the control signal to modify the input

signal to generate an output signal with compression and reduced noise.

Thus one of ordinary skill would have been motivated to seek a method of

reducing noise in an input in order to provide a method of an actual working

arrangement taught by "Sheikhzadeh, Sameti, Deng Brennan", Händel, and Eatwell in

combination. Such method would have been any known method for use in noise

enhancement such as one of Lindemann et al. in the same field of endeavor.

Lindemann et al. teaches (see Fig. 11) the steps of:

(6) supplying the input signal (LEFT IN) to an amplification unit (268; col.12, lines 62-67);

(7) providing the noise reduced signal (from 264) to a compression circuit which generates a control signal for the amplification unit (266) which generates a control signal for the amplification unit (268; col. 12, lines 12-61);

(8) controlling the amplification unit (268) with the control signal to modify the input signal to generate an output signal (LEFT OUT) with compression and reduced noise (col. 12, lines 62-67).

Therefore, it would have been obvious to one of ordinary skill in the art, at the time the invention was made to include within the "Sheikhzadeh, Sameti, Deng, Brennan" Händel, and Eatwell in combination a method of noise enhance system as taught by Lindemann et al. in order enhance desired sound and reduce undesired sound as suggested by Lindemann et al. in column 2, lines 51-53.

Regarding **claim 21,** this claim merely reflects the apparatus to the method claim of claim 2 and is therefore objected for the same reasons.

Regarding **claim 3,** "Sheikhzadeh, Sameti, Deng, Brennan" Händel, Eatwell, and Lindemann et al. in combination teaches a method as claimed in claim 2.   Lindemann et al. further teaches an original left and right inputs (FFT vectors) are multiplied (amplified) in operations 265, 267 by left gain and right gain vectors. The left gain and right gain vectors are frequency response adjustment vectors, which are specific to

each user and are a function of the audiogram measurements of hearing loss of the

user. These measurements would be taken during the fitting process for the hearing aid

(col. 12, lines 55-61). Thus one of ordinary skill would have been obvious to modify

"Sheikhzadeh, Sameti, Deng, Brennan" Händel, Eatwell in combination a method

subjecting the input signal to an auxiliary noise reduction algorithm to generate an

auxiliary noise reduced signal and providing the auxiliary noise reduced signal to the

compression circuit as taught by Lindemann et al. in column 12, lines 55-61.

Regarding **Claim 27,** this claim merely reflects the apparatus to the method claim

of claim 3 and is therefore objected for the same reasons.

Regarding **claim 26,** "Sheikhzadeh, Sameti, Deng, Brennan" in view of Händel,

in view of Eatwell, and further in view Lindemann et al teaches a method as claimed as

claim 2. "Sheikhzadeh, Sameti, Deng, Brennan" Händel, and Eatwell in combination

further teaches step (1) to (5) to the input signal prior to supplying the input signal to the

amplification unit.

Regarding **claim 22,** this claim merely reflects the apparatus to the method claim

of claim 1 and is therefore objected for the same reasons.

Regarding **claim 23,** "Sheikhzadeh, Sameti, Deng, Brennan" Händel, Eatwell,

and Lindemann et al. in combination teaches an apparatus as claimed in claim 22.

"Sheikhzadeh, Sameti, Deng, Brennan" further teaches a frequency transform means (FFT, Fig.1) connected between said input and both of the magnitude means and the spectral estimate means (Signal and noise spectrum estimates) for transforming the signal into the frequency domain to provide a transformed signal (X(f)) wherein the magnitude means determines the magnitude spectrum (IX(f)I) from the transformed signal (X(f)), and wherein the spectral estimate means determines the noise spectral estimate (| N | (f)) from the transformed signal (X(f)) in the absence of speech, the apparatus further including inverse frequency transform means (FFT $^{-1}$) for receiving a transformed noise reduced signal from the multiplication unit, the inverse frequency transform means providing the noise reduced signal (page I-13, left-hand column, paragraph 2 - right-hand column, paragraph 2).

Regarding **claim 24**, "Sheikhzadeh, Sameti, Deng, Brennan" Händel, Eatwell, and Lindemann et al. in combination discloses an apparatus as claimed in claim 23. "Sheikhzadeh, Sameti, Deng, Brennan" further teaches the noise filter calculation unit (Signal and noise spectrum estimates) determines the square of the speech magnitude spectral estimate by subtracting the square of the noise magnitude spectral estimate from the square of the magnitude spectrum of the input signal (page I-13, left-hand column, paragraph 2 - right-hand column, paragraph 2). However, "Sheikhzadeh, Sameti, Deng, Brennan" does not disclose the noise filter calculation unit calculates the attenuation function (H(f)), as a function of frequency. Händel further discloses a spectrum subtraction wherein (see page 7, Table 2, line 8) the attenuation function

means (24, Fig. 1) calculates the auxiliary signal as an attenuation function because the
method attenuation function with an oversubtraction factor and an attenuation rule gives
a better noise reduction without sacrificing audible quality (see page 2, lines 12-13).
Händel further discloses the equation:

$$H(f) = \{ (| X(f) |^2 - \beta | N^{\wedge}(f) |^2) / | X(f) |^2) \}^{\alpha}$$

Where: $H(f) = \hat{H}_{\delta PS}(w)$, attenuation function (see page 7, line 2-3)

$X(f) = \Phi^{\wedge}_x(w)$, magnitude spectrum of the input audio signal

(see page 3, line 26-27)

$N^{\wedge}(f) = \Phi^{\wedge}_v(w)$, noise magnitude spectral estimate

(see page 4, line 3-4)

β $= \delta$, oversubtraction factor (see page 26, lines 6-7)

α $= \frac{1}{2}$, an attenuation rule

Regarding **claim 4**, "Sheikhzadeh, Sameti, Deng, Brennan" Händel, Eatwell, and
Lindemann et al. in combination teaches a method as claimed in claim 3, wherein the
auxiliary noise reduction algorithm comprises the method of claim 1. Further, claim 4 is
interpreted and thus rejected for the reasons set forth above in the rejection of claim 1.

Regarding **claim 5**, "Sheikhzadeh, Sameti, Deng, Brennan" Händel, Eatwell, and
Lindemann et al. in combination teaches a method as claimed in claim 3. Lindemann et
al. further teaches a method wherein the auxiliary noise reduction algorithm is different

from the method of claim 1 (specific to each user, i.e., different parameter; col. 12, lines 55-61).

Regarding **claim 28,** this claim merely reflects the apparatus to the method claim of claim 4 and is therefore objected for the same reasons.

Regarding **Claim 29,** this claim merely reflects the apparatus to the method claim of claim 5 and is therefore objected for the same reasons

Regarding **claim 30,** "Sheikhzadeh, Sameti, Deng, Brennan" Händel, Eatwell, and Lindemann et al. in combination teaches a method as claimed as claim 22.

Eatwell further teaches a noise reduction filter in which the gains are adjusted according to estimates of the signal and noise contents of the signal components. When a signal component has a high signal-to-noise ratio, the gain should be set close to unity to allow the signal component to pass. When a signal component has a low signal-to-noise ratio, it is desirable to reduce the level of the component so as to reduce the noise in the output signal. Similar gain elements are used in the spectral subtraction method (col. 8, line 63 – col. 9, line 3) in order to provide a noise reduction filter with low computation and memory requirements (col. 3, lines 26-27).

Therefore, it would have been obvious to one of ordinary skill in the art, at the time the invention was made to include within the "Sheikhzadeh, Sameti, Deng, Brennan" in combination, a noise reduction filter to generate a noise reduced signal

using signal having only a high signal-to-noise ratio, and signal having only a high

signal-to-noise ratio; there is no substantial modification to the input signal for very low

and for very high signal to noise ratios; for the purpose of reducing the noise in the

output signal as taught by Eatwell in col. 8, line 63 – col. 9, line 3 in order to provide a

noise reduction filter with low computation and memory requirements as suggested by

Eatwell in column 3, lines 26-27.

4.   **Claims 15, 16** are rejected under 35 U.S.C. 103(a) as being unpatentable over

Sheikhzadeh, Sameti, Deng and Brennan ("Comparative Performance of Spectral

Subtraction and HMM-Based Speech Enhancement Strategies with Application to

hearing Aid Design", Acoustics, Speech, and Signal Processing, 1994. ICASSP-94. ,

1994 IEEE Int. Conf.) in view of Händel (Händel; PCT WO 96/24128), in view of

Eatwell U.S. Patent 5,742,694, in view of Nakajima et al. (U. S. Patent No. 4,283,601),

and further in view of Crepy et al. (4,924,508).

Regarding **claim 15**, "Sheikhzadeh, Sameti, Deng, Brennan", Händel, and

Eatwell in combination teaches a method as claimed in claim 14. Händel teaches a

method of reducing noise (see page 2, lines 10-12) in an input, audio signal containing

speech (see page 2, lines 3-6). The method includes detecting speech with a modified

auto-correlation. However, Händel does not disclose detecting speech with (partial)

auto-correlation function. In an analogous art, Nakajima et al. teach a method of

detecting speech (see Fig. 3 and Fig. 6) by using (partial) auto-correlation because the

synthesis of degrees of agreement of partial auto-correlation coefficients enhance the

filter stability (see col. 1, lines 57-59 and col. 10, lines 28-31). Nakajima et al. further

teach:

(1) taking an input sample and separating it into short blocks (160) and   storing the

blocks in correlation buffers (see col. 6, 42-43);

(2) correlating the blocks (160), to form partial correlations (111, see col. 6, lines

51-58); and

(3) summing the partial correlations(141,151) to obtain a final correlation see col.

6, lines 58-63).

Therefore, it would have been obvious to one of ordinary skill in the art, at the

time the invention was made to combines within Händel's method the (partial)

auto-correlation function method to detect the speech as taught by Nakajima in order to

enhance the filter stability (see col. 1, lines 57-59 and col. 10, lines 28-31). However

Nakajima does not explicitly shows correlating the blocks in parallel. Thus one of

ordinary skill would have been obvious to seek method the (partial) auto-correlation

function in order to work with the actual method of Nakajima. Such method would have

been any known method for use in auto-correlation such as one of Crepy et al. in the

same field of endeavor.

Crepy et al. teaches a method for use in auto-correlation for correlating the

blocks in parallel (Fig. 4; col. 6, lines 23-29) in order to provide an efficient method for

determining voice pitch related information (col. 1, lines 65-68).

Therefore, it would have been obvious to one of ordinary skill in the art, at the

time the invention was made to combine within Nakajima's method a (partial)

auto-correlation function method as taught by Nakajima in order to provide an efficient

method for determining voice pitch related information as suggested by Crepy et al. in

column 1, lines 65-68.

Regarding **claim 16**, both of Händel (see page 4 line 22- page 5 line 2) and

Nakajima (see col. 5 lines 9-13) teach methods of detecting speech directly in the

frequency domain by using digital signal processing with Fast Fourier Transform.

Nakajima et al. further teach a method of detecting speech (see Fig. 3 and Fig. 6) by

using Fast Fourier Transform (7, and see col. 5 lines 9-18) to generate partial

correlations (see col. 6 lines 28-36) because the synthesis of degrees of agreement of

partial auto-correlation coefficients enhance the filter stability (see col. 1, lines 57-59

and col. 10, lines 28-31).

5.   **Claims 17-20** are rejected under 35 U.S.C. 103(a) as being unpatentable over

Sheikhzadeh, Sameti, Deng and Brennan ("Comparative Performance of Spectral

Subtraction and HMM-Based Speech Enhancement Strategies with Application to

hearing Aid Design", Acoustics, Speech, and Signal Processing, 1994. ICASSP-94.,

1994 IEEE Int. Conf.) in view of Händel (Händel; PCT WO 96/24128), in view of

Eatwell U.S. Patent 5,742,694, in view of Yasunaga (U.S. Patent No. 4,845,753), and

further in view of Crepy et al. (4,924,508).

Regarding **claim 17**, "Sheikhzadeh, Sameti, Deng, Brennan", Händel, and

Eatwell in combination teaches a method as claimed in claim 1. However

"Sheikhzadeh, Sameti, Deng, Brennan", Händel, and Eatwell in combination does not

explicitly show the method wherein detecting the presence or absence of speech

comprises:

(1) taking a block of the input signal and performing an auto-correlation on that

block to form a correlated signal; and,

(2) checking the correlated signal for the presence of a periodic signal having a

pitch corresponding to that for a desired audio signal. Thus one of ordinary skill would

have been obvious to seek method in order to provide a method of an actual working

arrangement taught by "Sheikhzadeh, Sameti, Deng, Brennan", Händel, and Eatwell in

combination. Such method would have been any known method for use in noise

reduction such as one of Yasunaga in the same field of endeavor.

Yasunaga teaches a method of determining the presence of speech (Abstract,

line 6-12) in an audio signal (see Fig. 4). The method comprising taking a block of in

input audio signal (see col. 3 lines 39-40) and performing an auto-correlation (S42) on

that block to form a correlated signal (see col. 3 lines 43-45); and checking (S53) the

correlated signal for the presence of a periodic signal having a pitch corresponding to

that for speech (see col. 2 lines 58-61, col. 3 lines 65-67) in order to provide a pitch

detecting device in which the conventional drawbacks are removed and which has a

control means for controlling the order of an inverse filter in accordance with a mean

prediction residual obtained by spectrum data (col. 1, line 66 –col. 2, line 3).

Therefore, it would have been obvious to one of ordinary skill in the art, at the

time the invention was made to include within the "Sheikhzadeh, Sameti, Deng,

Brennan" Händel, and Eatwell in combination a method of noise enhance system as

taught by Yasunaga in order enhance desired sound and reduce undesired sound as

suggested by Yasunaga in column 1, line 66 –column 2, line 3.

However Yasunaga does not explicitly shows correlating the blocks in parallel.

Thus one of ordinary skill would have been obvious to seek method the (partial)

auto-correlation function in order to work with the actual method of Yasunaga. Such

method would have been any known method for use in auto-correlation such as one of

Crepy et al. in the same field of endeavor.

Crepy et al. teaches a method for use in auto-correlation for correlating the

blocks in parallel (Fig. 4; col. 6, lines 23-29) in order to provide an efficient method for

determining voice pitch related information (col. 1, lines 65-68).

Regarding **claim 18**, Yasunaga et al. further teaches a method of determining

the presence of speech (Abstract, line 6-12) in an audio signal (see Fig. 4), wherein the

auto-correlation (S42) is performed on a first block taken from an audio signal (see col.

3 lines 43-45), and a delayed block (S44) from the audio signal (Fig. 4; col. 6, lines 23-

29). Crepy et al. further teaches a method for use in auto-correlation for correlating the

blocks in parallel (Fig. 4; col. 6, lines 23-29)

Regarding **claim 19**, Yasunaga et al. teaches a method of determining the

presence of speech (Abstract, line 6-12) in an audio signal by performing

auto-correlation (see Fig. 4), wherein each block is subdivided into a plurality of shorter

sections (S43) and the correlation comprises correlation between pairs of the shorter

sections (S46) to form partial correlations (S48), and subsequently summing the partial

correlations to obtain the correlated signal (S52, and see col. 3 lines 40-67). Crepy et

al. further teaches a method for use in auto-correlation for correlating the blocks in

parallel (Fig. 4; col. 6, lines 23-29)

Regarding **claim 20**, Yasunaga teaches a method of determining the presence

of speech (Abstract, line 6-12) in an audio signal by performing partial correlation (see

Fig. 4), wherein an input signal is stored as a plurality of samples (S43) in a pair of

correlation buffers (S47), and the auto-correlation is performed on the signals in the

buffers to determine the partial correlations (S48), which partial correlations are

summed and stored (S52, and see col. 3 lines 40-67). Crepy et al. further teaches a

method for use in auto-correlation for correlating the blocks in parallel (Fig. 4; col. 6,

lines 23-29).

Allowable Subject Matter

6.   **Claims 7-11** objected to as being dependent upon a rejected base claim, but would be allowable if rewritten in independent form including all of the limitations of the base claim and any intervening claims.

Regarding to **claim 7**, the prior art provided numerous examples of different noise reduction methods but failed to disclose or fairly suggest the specific functional limitation as specify in claim 7, specifically the oversubtraction factor β is varied as a function of the signal to noise ratio, with β being zero for high and low signal to noise ratios and with β being increased as the signal to noise ratio increases above zero to a maximum value at a predetermined signal to noise ratio and for higher signal to noise ratios β decreases to zero at a second predetermined signal to noise ratio greater than the first predetermined signal to noise ratio.

Regarding to **claim 8**, the prior art provided numerous examples of different noise reduction methods but failed to disclose or fairly suggest the specific functional limitation as specify in claim 8, specifically the oversubtraction factor β is divided by a reemphasis function P(f) to give a modified oversubtraction factor $\beta^\wedge$ *(f)*, the preemphasis function being such as to reduce β at high frequencies, and thereby reduce attenuation at high frequencies.

Regarding **to claim 9**, the prior art provided numerous examples of different noise reduction methods but failed to disclose or fairly suggest the specific functional

limitation as specify in claim 9, specifically the rate of change of the attenuation function is controlled to prevent abrupt and rapid changes in the attenuation function.

Regarding to **claim 10**, the prior art provided numerous examples of different noise reduction methods but failed to disclose or fairly suggest the specific functional limitation as specify in claim 10, specifically the attenuation function is calculated at successive time frames, and the attenuation function is calculated in accordance with the following equation :

$G_n(f) = (1-\gamma)H(f) + \gamma G_{n-1}(f)$

Wherein $G_n(f)$ and $G_{n-1}(f)$ are the smoothed attenuation functions at the n'th and (n-1)'th time frames, and y is a forgetting factor.

Regarding to **claim 11**, the prior art provided numerous examples of different noise reduction methods but failed to disclose or fairly suggest the specific functional limitation as specify in claim 11, specifically β is the function of perceptual distortion.

Response to Arguments

7.   Applicant's arguments with respect to claims 1-6, and 12-30 have been considered but are moot in view of the new ground(s) of rejection.

Conclusion

8.   The following are suggested formats for either a Certificate of Mailing or Certificate of Transmission under 37 CFR 1.8(a). The certification may be included with all correspondence concerning this application or proceeding to establish a date of mailing or transmission under 37 CFR 1.8(a). Proper use of this procedure will result in such communication being considered as timely if the established date is within the required period for reply. The Certificate should be signed by the individual actually depositing or transmitting the correspondence or by an individual who, upon information and belief, expects the correspondence to be mailed or transmitted in the normal course of business by another no later than the date indicated.

Certificate of Mailing

I hereby certify that this correspondence is being deposited with the United States Postal Service with sufficient postage as first class mail in an envelope addressed to:

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I hereby certify that this correspondence is being facsimile transmitted to the United States Patent and Trademark Office, Fax No. (703)_____ - _____ on _____.
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Typed or printed name of person signing this certificate:

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Signature: _____

Please refer to 37 CFR 1.6(d) and 1.8(a)(2) for filing limitations concerning facsimile transmissions and mailing, respectively.

9.   Applicant's amendment necessitated the new ground(s) of rejection presented in this Office action. Accordingly, **THIS ACTION IS MADE FINAL**. See MPEP § 706.07(a). Applicant is reminded of the extension of time policy as set forth in 37 CFR 1.136(a).

A shortened statutory period for reply to this final action is set to expire THREE MONTHS from the mailing date of this action. In the event a first reply is filed within TWO MONTHS of the mailing date of this final action and the advisory action is not mailed until after the end of the THREE-MONTH shortened statutory period, then the shortened statutory period will expire on the date the advisory action is mailed, and any extension fee pursuant to 37 CFR 1.136(a) will be calculated from the mailing date of the advisory action. In no event, however, will the statutory period for reply expire later than SIX MONTHS from the date of this final action.

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Con P. Tran, whose telephone number is (703) 305-2341. The examiner can normally be reached on M - F (8:30 AM - 5:00 PM).

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Forester W. Isen can be reached on (703) 305-4386. The fax phone numbers for the organization where this application or proceeding is assigned are (703) 872-9314 for regular communications and (703) 872-9314 for After Final communications.

Any inquiry of a general nature or relating to the status of this application or

proceeding should be directed to the Customer Service Office at telephone number

(703) 306-0377.

cpt   CPT
February 28, 2003

FORESTER W. ISEN
SUPERVISORY PATENT EXAMINER
2600